

IN THE CLAIMS

1. (currently amended) ~~An~~ In an end-to-end estimation method of the bandwidth available in a ~~client-server~~ connection of client and server established over a packet switching network, the improvements comprising:
- computing in a first routine to compute samples of available bandwidth by taking into account ~~the a~~ flow of data packets received by the client ~~and in one RTT~~ time intervals during which the data packets are received if the routine is implemented at ~~the client a~~ receiver side of the client, or by taking into account acknowledgments or report packets received by a sender side of the server sender-side and in one RTT time intervals during which acknowledgment or report packets are received if the routine is implemented at the server sender side;
- computing in a second routine to compute samples of available bandwidth as ~~the a~~ ratio of ~~the an~~ amount of received data packets over the time interval during which the data packets are received if the routine is implemented at the client receiver side, or as ~~the a~~ ratio of ~~the an~~ amount of the data packets acknowledged over the time interval during which the data packets are acknowledged if the routine is implemented at the server sender side; and
- implementing in a third routine that implements a discrete time low-pass filter to obtain a filtered value of the samples of the available bandwidth.
2. (currently amended) The end-to-end bandwidth estimation method according to claim 1, wherein a sample of available bandwidth  $b_j$  at time  $t_j$  is computed as:

$$b_j = \frac{d_j}{t_j - t_{j-1}}$$

where  $d_j$  is the amount of the data packets that have been received at the client receiver side or acknowledged at the server sender side in the interval  $t_j - t_{j-1}$ ,  $t_{j-1}$  is the time when the previous ACK or the ACK one or more congestion windows of packets before were received by the server sender side or the a time when the a previous packet or the packet one or more congestion windows of packets before were received by the client receiver side, and  $t_j$  is the a time when the current ACK is received by the server sender side or when the a current packet is received by the client receiver side.

3. (currently amended) The end-to-end bandwidth estimation method according to claim 1, wherein the routine implements a discrete time low-pass filter with time-varying coefficients.

4. (previously presented) The end-to-end bandwidth estimation according to claim 2, wherein the available bandwidth samples are computed according to claim 2 and are averaged using the discrete-time low-pass filter with time-varying coefficients:

$$\hat{b}_j = \frac{2\tau_f - \Delta_j}{2\tau_f + \Delta_j} \hat{b}_{j-1} + \Delta_j \frac{b_j + b_{j-1}}{2\tau_f + \Delta_j}$$

where  $\hat{b}_j$  is the filtered measurement of the available bandwidth at time  $t = t_j$ ,  $\hat{b}_{j-1}$  is the filtered measurement of the available bandwidth at time  $t_{j-1}$ ,  $\Delta_j = t_j - t_{j-1}$ ,  $1/\tau_f$  is the cut-off frequency of the filter,  $b_j$  is the sample of the available bandwidth at time  $t_j$ , and  $b_{j-1}$  is the sample of the available bandwidth at time  $t_{j-1}$  if a time  $t/m$  ( $m \geq 2$ ) has elapsed since the last received ACK or packet without receiving any new ACK or packet, then the filter assumes

the reception of a virtual sample  $b_j=0$ .

5. (original) Method for adapting the amount of data for unit of time, i.e. the rate, sent by the server to the client over a packet network, comprising an end-to-end bandwidth estimation according to claim 1.

6. (previously presented) Method for adapting the amount of data for unit of time sent by the server to the client over a packet network, comprising an end-to-end bandwidth estimation comprising:

a routine to compute samples of available bandwidth by taking into account packets received by the client, if the routine is implemented at the receiver side, or by taking into account acknowledgment packets received by the sender, if the routine is implemented at the sender side; and

a routine that implements a discrete time low-pass filter to obtain a filtered value of the samples of available bandwidth,

wherein the low-pass filter is a low-pass filter according to claim 3.

7. (previously presented) Method for adaptively setting congestion window (cwnd) and slow start threshold (ssthresh) in the TCP/IP protocol comprising an end-to-end bandwidth estimation according to claim 1 to set the windows as follows:

after a timeout: set  $ssthresh = \min(2, BWE * RTT_{min})$

set  $cwnd = 2$ ;

after 3 dupack: set  $ssthresh = \min(2, BWE * RTT_{min})$

set  $cwnd = ssthresh$ ; and

wherein RTT min is the minimum round trip time and BWE is the available bandwidth computed according to claim 1 at the time of timeout or when 3 dupacks or n are received.

8. (previously presented) Method for adaptively setting congestion window and slow start threshold in the TCP/IP protocol comprising an end-to-end estimation of the bandwidth available according to claim 1.

9. (currently amended) Method for adaptively selecting the quality of coding, or the numbers of layers to be transmitted in a layered coding of an audio/video source using the TCP protocol, or the UDP protocol or the RTP protocol, comprising:

a routine to compute an end-to-end estimation of the available bandwidth according to claim 7;

a routine that selects the quality of coding, or the numbers of layers to be transmitted in a layered coding of an audio/video so that the sending rate is the closer to the end-to-end bandwidth available estimated according to claim 7; and

a routine to set TCP congestion window and slow start threshold according to claim 7 in order to send the coded audio/video source.

10. (previously presented) Method for adaptively selecting the quality of coding, or the numbers of layers to be transmitted in a layered coding of an audio/video source using the TCP protocol, or the UDP protocol or the RTP protocol comprising and end-to-end bandwidth estimation comprising:

a routine to compute samples of available bandwidth by taking into account packets received by the client, if the routine is implemented at the receiver side, or by taking into account acknowledgment packets received by the sender, if the routine is implemented at the sender side; and

a routine that implements a discrete time low-pass filter to obtain a filtered value of the samples of available bandwidth over a packet network, comprising an end-to-end bandwidth,

wherein the low pass filter is a low pass filter according to claim 3.

11. (previously presented) Method for adaptively selecting the quality of coding, or the numbers of layers to be transmitted in a layered coding of an audio/video source, comprising:

increasing step by step the quality of coding, or the numbers of layers to be transmitted in a layered coding of an audio/video source until congestion is experienced by means of control packets;

setting the quality of coding or select the numbers of layers to be transmitted after that a congestion episode is signaled by means of control packets in according with the available bandwidth estimation at time of congestion according to claim 1 and increasing again step by step the quality of coding or the numbers of layers to be transmitted in a layered coding to probe for extra available bandwidth until a congestion episode is experienced again.

12. (currently amended) Method for setting the an Advertised Window of TCP equal to ~~the~~ a minimum of the Advertised Window and the available bandwidth estimate times the minimum round trip time, wherein the available bandwidth estimate is computed according to claim 1.

13. (previously presented) Method for adapting the amount of data for unit of time sent by the server to the client over a packet network, comprising an end-to-end bandwidth estimation comprising:

a routine to compute samples of available bandwidth by taking into account packets received by the client, if the routine is implemented at the receiver side, or by taking into account acknowledgment packets received by the sender, if the routine is implemented at the sender side; and

a routine that implements a discrete time low-pass filter to obtain a filtered value of the samples of available bandwidth,

wherein the low pass filter is a low pass filter according to claim 4.

14. (previously presented) Method for adaptively setting congestion window and slow start threshold in the TCP/IP protocol comprising an end-to-end bandwidth estimation comprising:

a routine to compute samples of available bandwidth by taking into account packets received by the client, if the routine is implemented at the receiver side, or by taking into account acknowledgment packets received by the sender, if the routine is implemented at the sender side; and

a routine that implements a discrete time low-pass filter to obtain a filtered value of the samples of available bandwidth,

wherein the low-pass filter is a low-pass filter according to claim 4.

15. (previously presented) Method for adaptively selecting the quality of coding, or the numbers of layers to be transmitted in a layered coding of an audio/video source using the TCP protocol, or the UDP protocol or the RTP protocol comprising and end-to-end bandwidth estimation comprising:

a routine to compute samples of available bandwidth by taking into account packets received by the client, if the routine is implemented at the receiver side, or by taking into account acknowledgment packets received by the sender, if the routine is implemented at the sender side; and

a routine that implements a discrete time low-pass filter to obtain a filtered value of the samples of available bandwidth over a packet network, comprising an end-to-end bandwidth,

wherein the low-pass filter is a low-pass filter according to claim 4.

16. (previously presented) Method for adaptively selecting the quality of coding, or the numbers of layers to be transmitted in a layered coding of an audio/video source according to claim 10, comprising:

increasing step by step the quality of coding, or the numbers of layers to be transmitted in a layered coding of an audio/video source until congestion is experienced by means of control packets;

setting the quality of coding or select the numbers of layers to be transmitted after that a congestion episode is signaled by means of control packets in according with the bandwidth estimation comprising:

a routine to compute samples of available bandwidth by taking into account packets received by the client, if the routine is implemented at the receiver side, or by taking into account acknowledgment packets received by the sender, if the routine is implemented at the sender side;

a routine that implements a discrete time low-pass filter to obtain a filtered value of the samples of available bandwidth over a packet network, comprising an end-to-end bandwidth; and

increasing again step by step the quality of coding or the numbers of layers to be transmitted in a layered coding to probe for extra available bandwidth.

17. (currently amended) Method for setting the Advertised Window of TCP equal to the minimum of the Advertised Window and the available bandwidth estimate times the minimum round trip time, wherein the available bandwidth is computed according to claim 1.
18. (canceled)
19. (previously presented) Method for adapting the amount of data for unit of time, i.e. the rate, sent by the server to the client over a packet network, comprising an end-to-end bandwidth estimation according to claim 3.
20. (previously presented) Method for adapting the amount of data for unit of time, i.e. the rate, sent by the server to the client over a packet network, comprising an end-to-end bandwidth estimation according to claim 4.



21. (currently amended) The end-to-end bandwidth estimation method according to claim 2, wherein the routine implements a discrete time low-pass filter with time-varying coefficients.